• E-mail: Charles.jones125@yahoo.co.uk

SDR for the Revised Licence Exams

Tony Jones G7ETW follows up his earlier features on the revised exam syllabus with a closer look at those elements relating to digital signal processing (software defined radio).

Earlier in the year I wrote a series of articles detailing the changes to the Foundation, Intermediate and Full Licence Exams, which come into effect in September. By far the biggest change is the inclusion of the Digital Signals content, which students meet with increasing complexity at the three exam levels. I've split this content by level, explaining what needs to be taught. This article covers Foundation and Intermediate; a piece for the new Full Licence will follow. I include the actual syllabus points (verbatim: if it seems a bit repetitive, that's the syllabus, not me!)

Digital Signals at Foundation

Section 2F1: Digital vs Analogue signals

2F1 Syllabus points

• *Recall that analogue signals are constantly changing in amplitude, frequency or both.*

• *Recall that digital signals are a stream of finite values at a specific sampling rate.*

• *Recall that digital signals can be processed by a computing device with suitable software.*

See **Fig. 1**. This shows a single cycle of a 10Hz sinewave. It is constantly changing. If it were possible to infinitely zoom in on the curve, there would be a change in the y-axis (sine) value for even the smallest change in the x-value.

Fig. 2 shows the results of sampling this sinewave at 100Hz (one sample every 10ms). The resulting 'curve' is not constantly changing; the y-axis values rise and fall in a series of stepped values. Increasing the sampling rate to 250Hz, **Fig. 3**, increases the detail in the sample but the inherent granularity remains.

Sections 2F2 and 3M: Basic components of Digital signals operations

2F2 Syllabus points

• *Recall that an Analogue to Digital (ADC) converter is a device used to sample an analogue signal and produce a digital representation.*

• *Recall meaning of the term ADC.*

• *Recall that a computing device is required to process digital signals.*

• *Recall that a Digital to Analogue Converter is a device used to represent a digital signal in analogue format.*

• *Recall meaning of the term DAC.*

3M Syllabus points

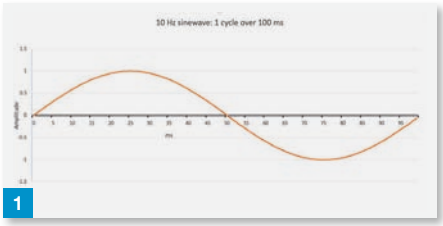
• *Recall that the SDR takes in all electromagnetic signals from the antenna and digitises this input for processing in software.*

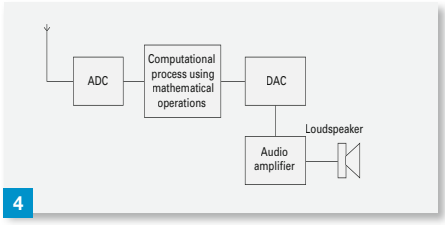
• *Recall that a mathematical operation enables all the signals to be sifted into separate frequency components.*

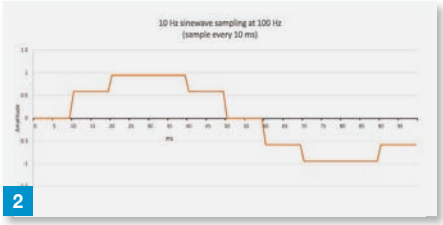
• *Recall that the required signal is selected using a filter defined in software.*

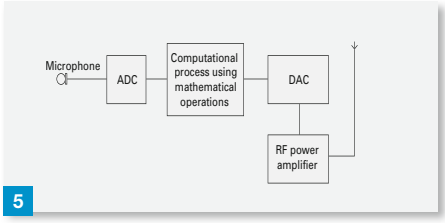
• *Recall that demodulation is carried out in software.*

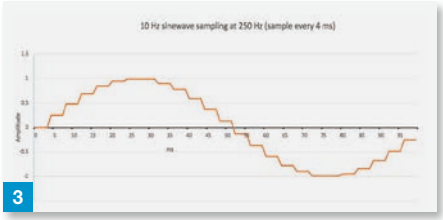
• *Recall that SDR receivers convert incoming signals into digital format and then perform filtering and demodulation on the signal using software and that SDR transmitters generate modulated radio signals using software.*











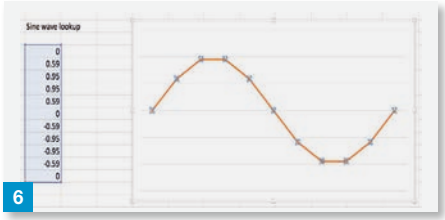


Fig. 1: 10Hz sinewave – one cycle takes 100ms. Fig. 2: The sinewave of Fig. 1 sampled every 10ms. Fig. 3: The sinewave of Fig. 1 sampled every 4ms. Fig. 4: SDR receiver block diagram (Foundation level). Fig. 5: SDR transmitter block diagram (Foundation). Fig. 6: 10-value lookup table created from Fig. 2.

SDR Receiver and Transmitter block diagrams

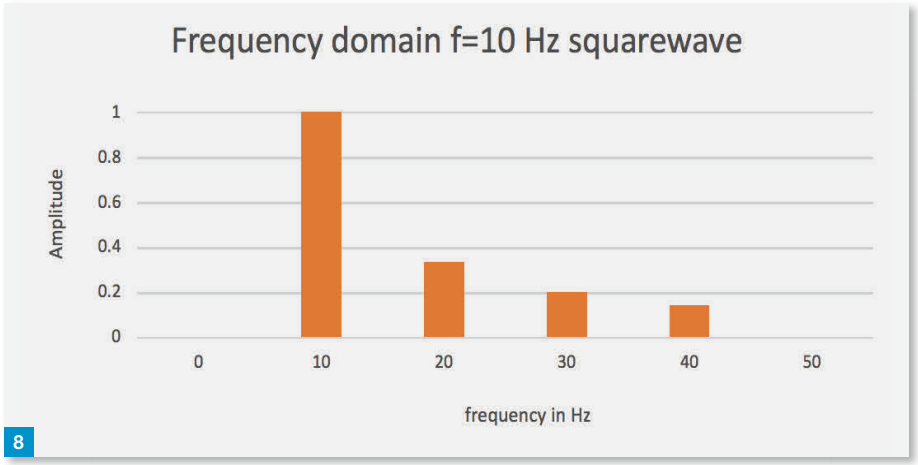
Fig. 4 is a block diagram of an SDR receiver.

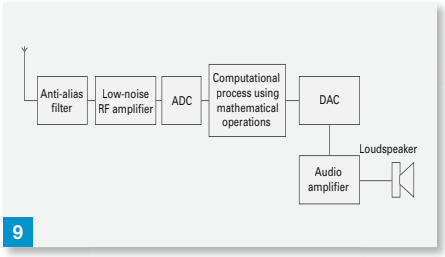
• ADC samples and digitises the incoming RF from the antenna.

• A computing device running software performs mathematical operations to sift the digitised RF into separate frequencies.

• Under user control, the computing device allows one signal to be selected using filters and digitally demodulated.

• DAC converts demodulated (but still digital AF) into an analogue AF waveform.





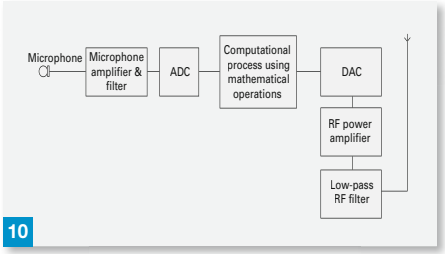


Fig. 7: A (rough) square-wave created by adding four sinewaves. Fig. 8: A frequency domain plot of the square wave in Fig. 7. Fig. 9: SDR receiver block diagram (Intermediate level). Fig. 10: SDR transmitter block diagram (Intermediate level).

• The AF waveform is amplified and can be heard through a loudspeaker.

Fig. 5 shows an SDR transmitter. This is conceptually the reverse.

• ADC samples and digitises the operator's voice.

• Computing device creates a digital RF frequency carrier and digitally modulates it creating a digital representation of a modulated RF signal.

• DAC converts the digital RF signal into an analogue RF waveform.

• The RF waveform is amplified and transmitted.

Digital Signals at Intermediate

Section 2F: Sampling

2F1 Syllabus points

• *Recall that digital signals with more bits and/or increased sampling rate enables a more accurate representation of the analogue signal.*

• *Recall that error introduced by sampling the analogue signal to produce the digital signal is a form of distortion.*

• *Recall the effect of increasing data rate on bandwidth requirements.*

• *Recall that the sampling rate needs to be at least twice the frequency of the analogue signal to adequately capture the detail of the analogue signal being sampled.*

• *Recall that the minimum sampling rate is called the Nyquist rate.*

It's ironic. People generally the word 'digital' to mean 'accurate'. It is a marketing person's go-to adjective, just as 'turbo' was in the 1980s. But digital sampling is an inherently distorting process, for two reasons:

Sampling causes distortion because, as shown earlier, all information in the input is lost between samples.

In addition, every sample adds distortion by the value being stored in binary. An Arduino Uno's ADC is a 100kHz, 10-bit sampler, meaning that input values (from 0V to 5V DC) are stored as numbers from 0 (binary 0000000000) to 1023 (1111111111). I am not going to explain binary storage of numbers. Please Google that if you need to.

5V digitised in 1023 steps makes the granularity or resolution of the system 4.888mV. Values that fall between these steps get rounded up or down depending on the digitisation software.

This is an unavoidable source of error. Using 16 bits per sample would reduce one bit's worth to 76µV. This allows much finer digitisation but this binary step distortion is truly unavoidable.

Nyquist rate, data rate bandwidth

There is a minimum sampling rate necessary for accurate processing, called the Nyquist rate. This is usually quoted as twice the highest frequency to be sampled, but that is incorrect. The Nyquist rate has to be more than that – say 2.1 times rather than twice this maximum frequency. CD stereo sound is sampled at 44.1kHz for a maximum frequency of 20kHz tops, more commonly 18kHz in practice.

The 'data rate' determines how much storage is needed for one second of sampling. It is the sampling rate multiplied by the sample size. For CD stereo sound this is 2 (channels) × 44,100 (samples per second) × 16 (bits per sample), which comes to 1.4Mb/s. (a lot.)

Data rate and bandwidth

As the data rate increases, this has a marked effect on 'bandwidth'. Please note the inverted commas there – bandwidth is a confusing term in this context.

My first thought was that in a digital communications context, 'bandwidth' means capacity of a data link. The greater the bandwidth, the greater the data rate and/or number of signals possible. With dial-up internet, running at the then-fabulous 28kb/s, only low-density data transfers could be handled. Now we have fibre-optic broadband, running at typically 35MB/s, and we can stream live TV. As

the data rate has gone up, so has the required bandwidth.

But there is also bandwidth's 'analogue' meaning: the frequency spread of a modulated carrier. Using a radio link, something must be done to modulate a carrier with digital data, and the faster the data rate, the faster this modulation must be. Any modulation creates sidebands, so as the modulating frequency goes up, the sidebands must get wider. As the data rate increases, so does RF bandwidth.

Recall the effect of increasing data rate on bandwidth requirements is all the syllabus says. I don't know which interpretation of bandwidth was meant, but fortunately the answer is the same: increasing the data rate increases the bandwidth required for digital signals.

Section 3C: DAC and Lookup tables

3C1 Syllabus points

- *Recall that digital signals can be used to generate audio and RF signals by Direct Digital Synthesis (DDS).*
- *Recall the meaning of DDS.*
- *Recall that a DDS generates audio and RF signals from pre-set values held in a memory or Look-Up table*.

See **Fig. 6**. This shows a 10-value look up table created from Fig. 2. The chart beside it shows this purely as a sequence of values. By controlling the timebase or period of this waveform, sinewaves (admittedly not very good ones) can be generated of any frequency. That is all that DDS is – a computer process that looks up values in a table and realises them as an analogue output.

Section 3M: SDR transmitters and receivers

3M1 Syllabus points

- *Recall that SDR software uses a mathematical function called a Fourier transform, which sifts the composite signal into its constituent independent frequencies for processing.*
- *Recall that this can also provide a spectrum or waterfall display.*
- *Recall that digital filters can be more effective than analogue filters.*

3M2 Syllabus points

- *Recall that meaning of the time domain and frequency domain.*
- *Understand how signals in the time domain can be viewed in the frequency domain.*
- *Identify some simple harmonic waves, the spectrum obtained using the Fourier transform.*

See **Fig. 7**. This shows a rough square wave I created in Excel by summing four related sinewaves. This is a 'time domain' plot; the y-axis is amplitude and the x-axis is time. It is not a very good sinewave deliberately – with a little imagination, the component sinewaves can be seen.

Foundation referred to a 'mathematical operation' that sifts the digitised RF into separate frequencies. Intermediate names this operation as 'Fourier Transform'.

Fig. 8 is a Time Domain plot and shows the Fourier Transform for this square wave. The y-axis is still amplitude but the x-axis is frequency. The four component sinewaves can be seen with their amplitudes.

The frequency domain shows that this square wave has a fundamental and three increasingly reduced levels of the third, fifth, seventh and so on harmonics.

Square wave = sin (x) + 0.33 sin (3x) + 0.2 sin (5x) + 0.14 sin (7x). This is precisely how I created it.

Waiter, bring me an Analogy

For people who prefer food to maths, consider a bowl of vegetable soup. Homogenous, murky and vaguely brown. Is that celeriac I can taste, or is it garlic? Turnip or swede? I pour the soup into my handy Fourier Transformatic, press the button and it gives me the recipe – 25% chicken stock, 20% carrots, 10% potatoes, 5% bacon, etc. With the recipe, I could recreate the soup.

Only waveforms with up to two harmonics will be examined.

Filtering

Analogue filters depend on the values of inductors and capacitors, and there is a limit to how accurately they can be built. Software filters are based on mathematics, implemented in computer code running on computing devices. Computer code does not have resistance, tolerances, or variance with temperature, and doesn't wear out. (At Intermediate, computer code is bug-free!)

This means that software filters can be created to tighter specifications than physical electronic ones and may outperform them.

SDR Receiver and Transmitter Block diagrams

3M3 syllabus points

- *Recall the different elements that make up the functions of an SDR (block diagram).*

See **Fig. 9** for an Intermediate SDR Receiver. This is more realistic than the diagram at Foundation level. It looks complicated but there is the same narrative symmetry to it. Students who understand the ADC – process – DAC cycle should have no problem understanding this.

In this diagram there is a box marked 'Low Pass Filter (anti-aliasing)'. 'Aliasing' is covered by Section 2F1 of the Advanced syllabus but is not included in the Intermediate syllabus. Tutors will therefore need to cover this at a bare minimum level. The next two paragraphs will suffice.

Any frequencies in an incoming waveform that are higher than half of the sampling rate cause problems (and add to distortion) because the sampling process makes them appear as related frequencies *below* half of the sampling rate. These false lower frequencies are called aliases.

To stop this happening, 'anti-aliasing' lowpass filters are used to ensure that the incoming waveform simply has no component frequencies above half the sampling rate, preferably with a safe margin for safety (hence the practical limit of 18kHz on CD sound).

Fig. 10 shows an Intermediate SDR Transmitter. It's worth explaining that in the computing process any kind of RF signal can be created, but its 'output' is a digital representation of what is required. An antenna (and an RF amplifier) is an analogue thing and wants an analogue AC current and voltage at radio frequencies as its input, which is what the DAC is there for.

Notes on Block Diagrams (both syllabuses)

'Official' diagrams for the Foundation SDR receiver and transmitter can be found in Table 2 of the new syllabus document below but Section 3M does not specifically say that these are examinable! We have to assume they are.
https://tinyurl.com/y2bpy6u6

Intermediate Syllabus Section 3M3 refers to an SDR radio block diagram so this definitely is examinable, but no diagram is present in Table 3 of the above document, where it logically should be. In the new Intermediate book by **Smart**, **Mills** and **Bleaney** there is a block diagram of an SDR transceiver but I have not used it because it loses the consistent left-to-right input-to-output flow that makes this easy to teach.